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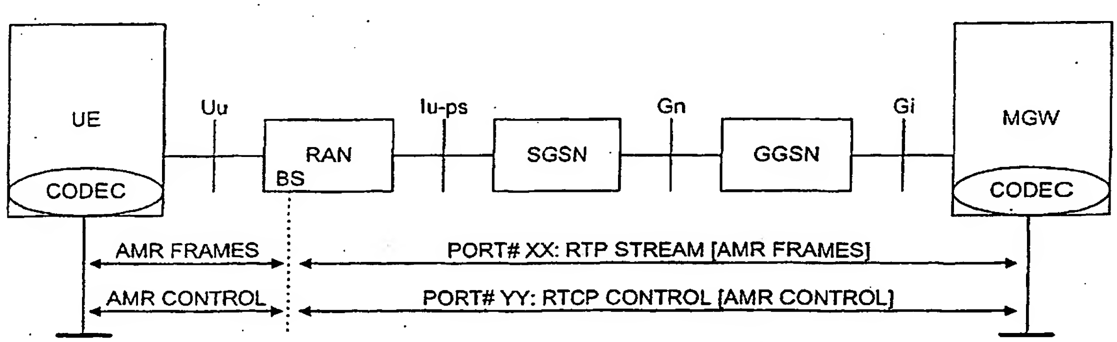
(54) Title: APPLICATION OF RTP AND RTCP IN THE AMR TRANSPORT IN VOICE OVER IP NETWORKS



(57) Abstract: The invention proposes a method for conveying an Adaptive Multi-Rate (AMR) codec stream between a first codex and a second codec over a network, comprising the steps of transmitting AMR frames via a Real-Time Transport Protocol (RTP) stream, and transmitting an AMR control signal via a Real-Time Transport Control Protocol (RTCP) stream. Thus, an easy control of the AMR codec stream via UMTS or IP networks is possible.

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APPLICATION OF RTP AND RTCP IN THE AMR TRANSPORT IN VOICE OVER IP NETWORKS

Field of the invention

The present invention relates to a method and a network system for conveying an Adaptive Multi-Rate (AMR) codec stream over multimedia IP networks.

BACKGROUND OF THE INVENTION

The invention relates to transmitting an Adaptive Multi-Rate (AMR) codec stream over networks. In particular, such a network can be a multimedia IP network, a third generation (3G) mobile network or a VoIP (Voice over IP) network in general. For UMTS (universal Mobile Telecommunication System) in 3GPP (3rd Generation Partnership Project), an AMR speech coding scheme has been selected. It is also to be used by the GSM (GPRS/EDGE GSM). The UMTS Release 2000 is targeted to provide IP service support over the UMTS air interface (Uu).

Usually, e.g. in GSM, a transmission channel between two codecs (coder-decoder) in a network has a fixed data rate. In response to certain conditions of the channel, e.g., the connection quality or depending on the data rate of the source, however, it is advantageous to change the channel data rate. This changing is performed by using AMR.

In an implementation, for example, two mobile stations are connected with each other via a mobile network. The first mobile station contains a first codec, and the second mobile stations contains a second codec. The codecs perform the necessary encoding/decoding for converting a voice signal into a digital signal which is transmitted via the network and vice versa. The codecs serve to provide a certain data rate, for example, 22.8 kbit/s on a so-called "full rate channel". In case of AMR codecs, it is possible to switch this data rate to another data rate, for example, to 11.4 kbit/s on a so-called "half rate channel". This switching between the different data rates has to be performed simultaneously by both of the codecs involved.

Therefore, a control of the AMR codecs involved is important. At present (UMTS release 1999), the control of the AMR codec is UMTS specific (ref. e.g., to 3GPP TS25.415).

However, such a control is not sufficient, in particular in case besides of UMTS also other network types are involved.

SUMMARY OF THE INVENTION

Accordingly, the object underlying the present invention resides in providing a method by which an improved control of AMR codecs is possible.

This object is solved by a method for conveying an Adaptive Multi-Rate (AMR) codec stream between a first codec and a second codec over a network, comprising the steps of transmitting AMR frames via a Real-Time

Transport Protocol (RTP) stream, and transmitting an AMR control signal via a Real-Time Transport Control Protocol (RTCP) stream.

By this method, i.e., using RTP and RTCP, a control channel is provided between the two peer AMR codecs in order to benefit from the capabilities of the AMR concept. Thus, the codecs have a mechanism to signal to its peer the requests to change the codec mode.

Furthermore, in case of a WCDMA (Wideband CDMA, Wideband Code Division Multiple Access) interface, the macrodiversity needs to be taken into account in the AMR control, as all branches of the call need to have the same coding mode. Since according to the invention the coding mode, i.e., the AMR frames, are the same during the transmission even in different networks, this condition can be fulfilled by the invention.

Furthermore, by the method according to the invention, the interfaces are re-usable between different application areas. Thus, the number of different interfaces (protocol stack, procedures) in UMTS or in a Voice over IP environment (wireless and/or fixed) can be minimized in order to avoid interworking problems and unnecessary implementation effort.

A first network element, in which the first codec may be arranged, can be a mobile station, the method further comprising the step of performing a handover from a first base station to a second base station, wherein each base station is adapted to determine a need for a codec mode change and to suggest a codec mode independently and the mobile station is adapted to decide of the codec mode change on the basis of a predetermined criterion.

The predetermined criterion can be using the lower quality mode of the codec modes suggested by the base stations. By this measure, a secure connection can be achieved.

Alternatively, the predetermined criterion can be using the higher quality mode of the codec modes suggested by the base stations. By this measure, a high quality connection can be achieved.

The first network element can be a mobile station, the method further comprising the step of performing a handover from a first base station to a second base station, wherein a network control element is adapted to determine a codec mode change. By this measure, a central network element (e.g., a Serving Radio Network Controller SRNC) is used for deciding on the codec mode. In this way, the mobile stations do not need to have the function of deciding on the codec mode. Such, the operational load for the mobile stations is reduced.

A codec mode which has been determined by the network control element as described above can be sent to the first base station and/or the second base station via the Real-Time Transport Control Protocol stream.

The invention also proposes a network element comprising a frames extracting/inserting means which is adapted to insert and/or extract Adaptive Multi-Rate (AMR) frames from and/or into a Real-Time Transport Protocol (RTP) stream, and a control signal extracting/inserting means which is adapted to insert and/or extract an AMR control signal from and/or into a Real-Time Transport Control Protocol (RTP) stream.

The network element can be a mobile station, a gateway or the like.

The network element can comprise a mode deciding means adapted to determine a need for a codec mode change and to suggest a codec mode.

The network element can further comprise a mode deciding means adapted to decide, in case of a handover between a first base station and a second base station, a codec mode change on the basis of received suggested codec mode changes and of a predetermined criterion.

The predetermined criterion can be using the lower quality mode of the codec modes suggested by the base stations, or, alternatively, using the higher quality mode of the codec modes suggested by the base stations.

The mode deciding means can be arranged in a further network element (SRNC).

Alternatively, the above object is solved by a network system comprising a first network element including a first codec for generating Adaptive Multi-Rate (AMR) codec frames, a second network element (BS) for receiving said AMR frames and including a transmitting means (CCU) for transmitting said AMR frames via a Real-Time Transport Protocol (RTP) stream and for transmitting an AMR control signal via a Real-Time Transport Control Protocol (RTCP) stream via the network, a third network element (GW) for receiving said RTP stream and said RTCP stream, for extracting said AMR frames from said RTP stream, for extracting said AMR control signal from said RTCP stream and for outputting said AMR frames and said

AMR control signal, and a fourth network element for receiving and processing said AMR frames and said AMR control signal.

The first and the second network elements can be separately arranged network elements, or, alternatively arranged within a single network element.

The third and fourth network elements can be separately arranged network elements or, alternatively, be arranged in a single network element.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be more readily understood with reference to the accompanying drawings in which:

Fig. 1 shows a block diagram of a network configuration in which a method according to an embodiment is applied;

Fig. 2 shows a protocol stack for the transport within the network shown in Fig. 1;

Fig. 3 shows a practical implementation of the embodiment shown in Fig. 1;

Fig. 4 shows the situation of a handover between two base stations according to a second embodiment;

Fig. 5 shows the situation of a handover between two base stations according to a third embodiment; and

Fig. 6 shows an example for a gateway without a codec.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

In the following, a preferred embodiment of the invention is described in more detail with reference to the accompanying drawings.

However, firstly, the idea according to the invention is described in general with reference to a block diagram shown in Fig. 1.

The method according to the invention uses the so-called Real-time Transport Protocol (RTP) to transmit AMR frames via a network, and the Real-time Transport Control Protocol (RTCP) is applied to transmit corresponding AMR control commands necessary for the codecs involved via the network.

The Real-time Transport Protocol (RTP, RFC1889) is a widely-used transport protocol for any real time communications over IP. There are already many payload types defined for RTP in order to support specific real time applications (like H.261 video, G.721 audio, etc.).

An integral part of the RTP is the Real-Time Control Protocol (RTCP). It provides the control channel for the RTP data sources. The RTCP stream is separated from the associated RTP data stream by the port number (e.g., UDP port number, i.e., the User Datagram Protocol port number). From the IP point of view, the control can be considered in-band.

The invention is to apply RTCP for end-to-end codec control (i.e., control between the peer codecs) in UMTS and in whatever environment the AMR is used. It is noted that a narrow-band or a wide-band AMR can be used.

As a consequence, there is a need to specify an RTP payload type for AMR coded speech as well as the mode change support into RTCP.

The method is described in the following with respect to Fig. 1.

On the left side, a User Equipment (UE) is shown which comprises an AMR codec (coder-decoder) for converting source signals like speech signals into data signals having a particular data rate. The AMR codec is adapted to form the output data signals into AMR frames, i.e., to transmit these data signals in an AMR frame stream or AMR frame channel.

The necessary control signals for signalling changes of the codec mode and so on to a counterpart codec on the receiving side, i.e., the AMR control signals are transmitted via an AMR control stream or AMR control channel. It is noted that the AMR control signals include AMR mode commands. It is noted that the counterpart codec resides in this case in a Media Gateway (MGW). That is, the MGW may contain the codec function (physically the transcoder). In general, the MGW performs functions like media conversions from one format to another (e.g., from an IP network to a Public Switched Telephone Network (PSTN)), payload processing (speech transcoding (i.e., the function of a codec), echo cancelling etc.).

The AMR frames and the AMR control signals are transmitted via the UMTS Uu interface to the Radio Access Network (RAN). In particular, the AMR frames and the AMR control signals are received by a Base Station (BS). In the RAN, the AMR frames which have been transmitted from

the UE to the BS are inserted into an RTP stream, and the AMR control signals which have been transmitted from the UE to the BS are inserted into an RTCP control stream.

Between the RAN and the UMTS Core Network, the RTP stream and the RTCP stream are conveyed via an Iu interface. The Iu interface can be the Iu-PS interface between the RAN and the Packet Switched (PS) Core Network (CN). However, it is noted that the invention is also applicable for the Iu-CS interface between a Circuit Switched (CS) Core Network (CN).

The RTP stream and the RTCP stream are transmitted via an interface Iu-PS (also defined in UMTS) to a Serving GPRS Support Node (SGSN). Both streams are transmitted via an interface Gn (also defined in UMTS) to a Gateway GPRS Support Node (GGSN). Finally, the RTP stream and the RTCP stream are transmitted via an interface Gi to a Gateway (MGW) which comprises a codec. The gateway is adapted to extract the AMR frames from the RTP stream and to extract the AMR control signals from the RTCP control stream. The codec of the gateway converts the received AMR frames into the data signals as needed for further transmitting or processing in the network to which the gateway is connected. Furthermore, the codec is adapted to change its mode in response to the AMR control signals received in the RTCP control stream.

In Fig. 2, the corresponding protocol stack for the transport outlined above is shown. It is noted that this protocol stack is to be defined for the UMTS Release 2000 (R00) terrestrial transport, i.e., the transport between the Base Station and the GW node (i.e., the node where the codec resides in the network side).

The user data part of the stack presented in Fig. 2 can be e.g., AMR speech frames conveyed in a header compressed RTP/UDP/IP payload. The header compression/stripping has been standardized in UMTS to take place in the Mobile Station (MS, i.e., the UE) and in a Radio Network Controller (RNC, as a part of the RAN).

In the following, a practical implementation is described with respect to Fig. 3.

Here, the UE as shown in Fig. 1 is a Mobile Station MS. The mobile station MS comprises a codec including a Channel Coding Unit CCU to be described later. The MS is connected via a radio connection with a Base Station BS. The base station BS also comprises a Channel Coding Unit CCU.

Here, the CCU serves as a frames and control signal inserting/extracting means. In particular, an AMR frame is inserted into the RTP frame payload as such. In the Base Station, the Channel Coding Unit CCU performs the radio channel coding (downlink) and decoding (uplink). That is, in the case of an uplink, the CCU receives the AMR frame and the codec mode information (i.e., AMR control signal) for the uplink radio frame structure which has been transmitted from the UE to the BS. It then passes this information to a RTP protocol entity and to an associated RTCP protocol entity, respectively.

The base station BS is connected to a Radio Network Controller (RNC). The base station BS and the RNC are parts of the RAN (Radio Access Network) shown in Fig. 1. The RNC is connected with a Gateway GW which comprises a further codec. It is assumed that the AMR frames

transmitted between the mobile station MS and the gateway GW and vice versa are speech frames. As a matter of course, also other kind of traffic can be transmitted, for example video data and other data which require a real-time treatment or have to be delivered quickly for other reasons.

Furthermore, the Channel Coding Units (CCU), which are included in the MS and the BS, serve to determine the radio interface conditions, the resulting frame quality and the like, and serve to set different codec modes on the basis of these determination results. The CCU performs, e.g., the convolutional coding/decoding or Turbo coding/decoding for the radio interface transmission. That is, the CCUs set the corresponding AMR codec mode commands into the AMR control signals. In the BS, the CCU inserts the commands, i.e., the AMR mode commands, into the RTCP stream.

Now, a case is described in which the codec mode is to be changed.

The AMR codec change command for the downlink (DL) AMR stream (CHANGE DL MODE) is received over the radio interface, i.e., the interface between the MS and the BS. In Fig. 3, such a transmitting of a change command for the downlink AMR stream is indicated by the dotted arrow directed from the MS to the BS and further to the GW. The receiving BS inserts the information into the RTCP message that it subsequently sends towards the network node where the codec resides (which is in Fig. 3 the GW). As soon as the GW receives the CHANGE DL MODE command inserted into the RTCP message, the codec mode is changed accordingly (the requested mode is included in the

command). Each AMR frame carries the information of the used codec mode.

The change of codec mode is generally based on the radio interface conditions and the resulting frame quality that is determined by the Channel Coding Unit (CCU) located in the base station BS (UL) and in the mobile station MS (DL). In particular, with respect to the uplink (UL), the CCU of the base station BS performs such a determination, and concerning the downlink (DL), the mobile station MS performs the determination.

In the uplink (UL) AMR stream, the change command only needs to be sent over the radio interface, provided that there is only one base station participating in the call. In Fig. 3, the change command for the uplink AMR stream (CHANGE UL MODE) is indicated by a dotted arrow directed from the base station BS to the mobile station MS.

If there are more than one base station involved (e.g., due to soft hand-over) then the mechanism for UL codec mode change may be different. In the following, two alternative procedures are described as a second and a third embodiment, respectively.

Fig. 4 shows a situation, in which a (soft) handover between two base stations BS1 and BS2 according to the second embodiment is illustrated. As in the first embodiment, the mobile station MS comprises a codec and a CCU, and the base stations BS1 and BS2 each comprise a CCU. For simplifying the description, the remaining parts of the network (i.e., RNC, GW and the like) are omitted.

According to the second embodiment, each base station involved in the given AMR call determines the need for UL codec mode change independently from each other.

They also signal the request independently in the L1 frame over the radio interface. In Fig. 3, this is indicated by the dotted arrows directed from the base station BS1 to the MS and directed from the base station BS2 to the MS, respectively. Each of the base stations BS1 and BS2 sends a codec change command. Thus, the mobile station MS receives from the base station BS1 a command CHANGE UL MODE1 and from the base station a command CHANGE UL MODE2.

The MS is assigned the responsibility to make the decision of the mode change according to predefined criteria. In particular, the CCU of the MS is adapted to decide which one of the received codec change command is to be applied by the codec.

For example, it is assumed that a handover from the base station BS1 to the base station BS2 is effected. Thus, the current codec mode is defined by the base station BS1. At the time of the handover, the base station BS2 requests another codec mode by issuing the command CHANGE UL MODE2 via the L1 layer. Now, the mobile station MS has to decide which one of the modes should be applied, that is, whether the current mode should be changed to the new mode as requested by the base station BS2, or whether the current codec mode should be maintained.

A criterion can be, e.g., to apply the current mode if it is better than the proposed new mode. That is, the CCU of the MS selects the codec mode which provides a better quality. For example, in case the current codec mode is

full rate and the new mode requested by the base station BS2 is half rate, the CCU decides to maintain the full rate since the full rate provides a better quality.

This could lead to the case that the handover is not carried out, and even to dropping the corresponding branch from the call due to unacceptably quality.

Hence, this criterion should be applied in case a high quality connection is absolutely necessary.

An alternative criterion can be to apply the lower quality mode of the two. That is, the CCU of the MS selects the codec mode which has the lowest data rate. For example, in case the current codec mode is full rate and the new mode requested by the base station BS2 is half rate, the CCU decides to apply the half rate since the half rate has the lowest quality.

Thus, this criterion aims for keeping the branch active, at the cost of overall quality of the call.

As a matter of course, further criteria are possible. For example, in case an application in the mobile station MS requires a certain data rate, a criterion can be to apply that codec mode which provides this certain data rate.

Next, the second alternative procedure for handling a handover is described as the third embodiment.

According to the third embodiment, the UL codec mode change is determined by the Serving RNC (SRNC) of the calls by some defined algorithm. In this case, the CHANGE UL MODE command needs to be conveyed in the RTCP message (i.e., within the RTCP stream) to the SRNC from the base

station that requests the codec mode change. Then the decision of the change is signaled via RTCP back to all involved base stations.

This principle is illustrated in Fig. 5. As in Fig. 4, the situation is shown in which a handover from the base station BS1 to the base station BS2 takes place. In this case, however, the mobile station MS is not involved in the decision on which codec mode is to be applied. Instead, the CCU of the base station BS1 and the CCU of the base station BS2 transmit requests for a change of the UL mode to a SRNC (Serving RNC). The requests are in the form of AMR control signals which are transmitted via RTCP. The AMR control signal contain also the necessary information (e.g., radio interface conditions and the like). For simplifying the illustration, the RTP streams are omitted and only the RTCP streams are shown.

The SRNC performs a mode decision, for example by use of a particular control unit on the basis of the received control signals. Thereafter, the SRNC inserts the decided mode, which is an AMR control signal (AMR mode command) into the RTCP stream by which the decided codec mode is transmitted to the base stations BS1 and BS2. In turn, the base stations BS1 and BS2 transmits the decided codec mode (i.e., the decided UL mode) to the mobile station MS.

Generally the reason for the codec mode change can be something else than the radio interface conditions. It can be used e.g., for congestion avoidance. In this case the same principles apply as have been defined in the above text. That is, the RTCP is used as the control channel for indicating the need for the codec mode change.

It is noted that in the embodiments described above, the RTP and RTCP protocols do not have to be terminated in the network element where the codecs are located. They can be applied also only in some parts of the route between the first codec and the second codec. In Fig. 3, for example, the codecs are in the UE (i.e., the MS) and in the GW, whereas the RTP/TCP is used only between the BS and the GW.

In case of UMTS, the idea, as it has been illustrated in Fig. 1, is that in the User Equipment (Mobile Station) there does not have to be RTP and RTCP protocols implemented. Instead, the speech frames generated by the codec in UE can be transmitted over the radio interface (Uu) by whatever means. In any case the command to change the downlink codec mode (change-DL-mode command) is sent. This is a normal AMR functionality. This information may be conveyed, e.g., in the radio frame structure itself (in Layer 1 frame, no additional protocols involved). The implementation according to the invention takes this change-DL-mode information from the received radio frame (in the Base Station).

In realistic cases, the involved entity is the Channel Coding Unit (CCU) or an equivalent, that performs the radio channel decoding in the receiver. Then, only after the CCU, the RTP and RTCP are introduced. After having decoded the speech frame and the change-DL-mode information from the cannel coded radio frame (the radio frame may have been convolutionally coded, for example) the CCU passes this information to RTP (speech frame) and to RTCP (change mode information). After that, the RTP/RTCP convey this information. As soon as the RTP becomes terminated (either in the second node, i.e., the

destiny, or before that in some Gateway or equivalent), the information needs to be taken out from RTP/RTCP and re-packed accordingly. For example, if there was an MS-MS call and a Transcoder Free Operation (TrFO) was applied, then the RTP/RTCP stream would become terminated in the second BS and its CCU would take the change-DL-mode information and insert it into the radio frame along with the speech frame (from RTP payload).

Fig. 6 shows a further example in which the Gateway GW on the left side performs only the functionality of extracting and inserting of AMR frames into the RTP stream and the functionality of extracting and inserting AMR control signals in the RTCP stream. That is, this Gateway is an example for a network element in which no codec is included and which does not perform any decision regarding the codec mode or the like. This gateway merely extracts and/or inserts AMR frames into or from the RTP stream and extracts and/or inserts AMR control signals into or from the RTCP stream.

The RTP and RTCP streams are conveyed via an IP network (i.e., an "IP island") to a Media Gateway MGW which comprises a codec. In the codec, the original signal (e.g., a voice signal) is generated and transmitted via a, e.g., a circuit switched network to an opposite communication party.

The above description and accompanying drawings only illustrate the present invention by way of example. Thus, the embodiments of the invention and the modifications thereof described may vary within the scope of the attached claims.

In particular, it is noted that the embodiments can be arbitrarily combined. For example, the second and the third embodiment can be combined. This can be effected with respect to the type of the mobile station concerned. That is, if the mobile station is capable of determining and suggesting an appropriate AMR codec mode, and if it also is capable to decide between different codec modes which are suggested by base stations involved in a handover, the method according to the second embodiment can be applied. Thus, only in case the mobile stations concerned are not capable of performing such determinations and decisions, the method according to the third embodiment can be applied. That is, only in this case a central network element performs the decision.

Furthermore, the above embodiments have been described with respect to a structure in which a codec is arranged in a mobile station, and a further codec is arranged in a gateway. As a matter of course, this structure can be varied arbitrarily. For example, the gateway can be replaced by a second mobile station which is connected to the network via a second base station.

Claims

1. A method for conveying an Adaptive Multi-Rate (AMR) codec stream between a first codec and a second codec over a network, comprising the steps of

transmitting AMR frames via a Real-Time Transport Protocol (RTP) stream, and

transmitting an AMR control signal via a Real-Time Transport Control Protocol (RTCP) stream.

2. The method according to claim 1, wherein the first codec is arranged in a first network element (MS), and the second codec is arranged in a second network element (GW).

3. The method according to claim 2, wherein the first network element (MS) is a mobile station, the method further comprising the step of

performing a handover from a first base station (BS1) to a second base station (BS2), wherein each base station is adapted to determine a need for a codec mode change and to suggest a codec mode independently and the mobile station (MS) is adapted to decide of the codec mode change on the basis of a predetermined criterion.

4. The method according to claim 3, wherein the predetermined criterion is using the lower quality mode of the codec modes suggested by the base stations.

5. The method according to claim 3, wherein the predetermined criterion is using the higher quality mode of the codec modes suggested by the base stations.

6. The method according to claim 2, wherein the first network element (MS) is a mobile station, the method further comprising the step of

performing a handover from a first base station (BS1) to a second base station (BS2), wherein a network control element (SRNC) is adapted to determine a codec mode change.

7. The method according to claim 2, wherein a determined codec mode is sent to said first base station (BS1) and/or said second base station (BS2) via said Real-Time Transport Control Protocol stream.

8. A network element comprising

a frames extracting/inserting means which is adapted to insert and/or extract Adaptive Multi-Rate (AMR) frames from and/or into a Real-Time Transport Protocol (RTP) stream, and

a control signal extracting/inserting means which is adapted to insert and/or extract an AMR control signal from and/or into a Real-Time Transport Control Protocol (RTP) stream.

9. The network element according to claim 8, wherein said network element is a mobile station (MS).

10. The network element according to claim 8, wherein said network element is a gateway (GW).

11. The network element according to claim 8, further comprising a codec adapted to generate and/or process said AMR codec stream and said AMR control signal.

12. The network element according to claim 8, further comprising a mode deciding means (CCU) adapted to

determine a need for a codec mode change and to suggest a codec mode.

13. The network element according to claim 9, further comprising a mode deciding means (CCU) adapted to decide, in case of a handover between a first base station (BS1) and a second base station (BS2), a codec mode change on the basis of received suggested codec mode changes and of a predetermined criterion.

14. The network element according to claim 13, wherein the predetermined criterion is using the lower quality mode of the suggested codec modes.

15. The network element according to claim 13, wherein the predetermined criterion is using the higher quality mode of the suggested codec modes.

16. The network element according to claim 12, wherein said mode deciding means is arranged in a network control element (SRNC).

17. The network element according to claim 16, wherein said mode deciding means transmits a determined codec mode to said first base station (BS1) and/or said second base station (BS2) via said Real-Time Transport Control Protocol stream.

18. A network system comprising

a first network element (MS) including a first codec for generating Adaptive Multi-Rate (AMR) codec frames;

a second network element (BS) for receiving said AMR frames and including a transmitting means (CCU) for transmitting said AMR frames via a Real-Time Transport Protocol (RTP) stream and for transmitting an AMR control

signal via a Real-Time Transport Control Protocol (RTCP) stream via the network;

a third network element (GW) for receiving said RTP stream and said RTCP stream, for extracting said AMR frames from said RTP stream, for extracting said AMR control signal from said RTCP stream and for outputting said AMR frames and said AMR control signal; and

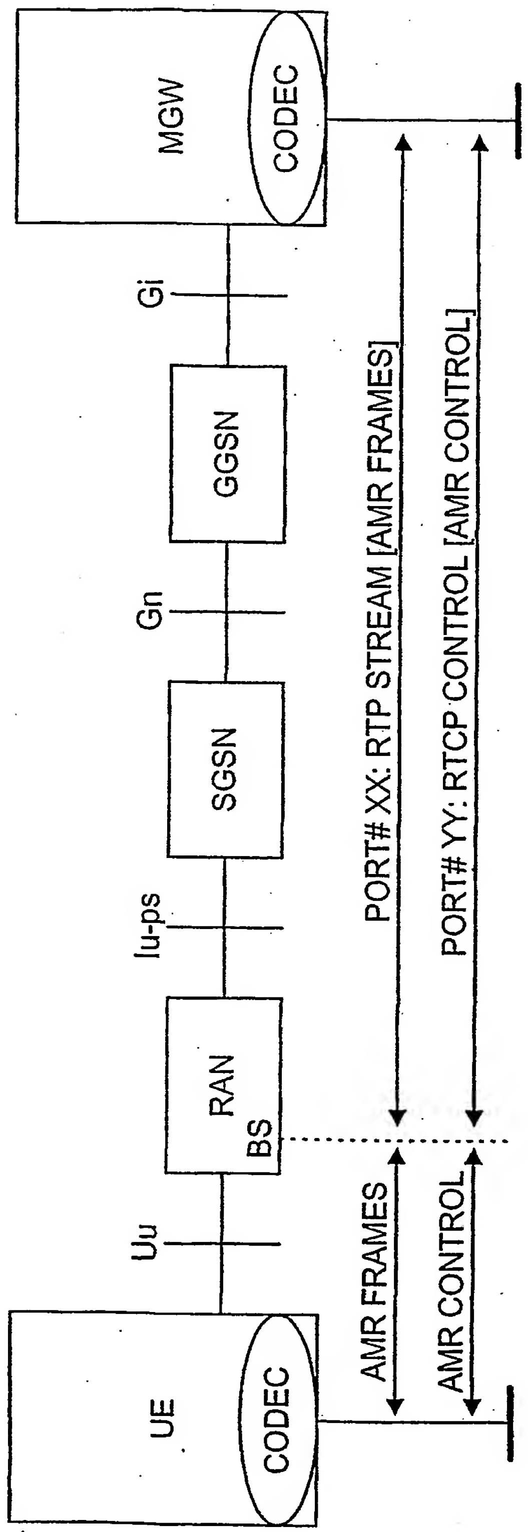
a fourth network element for receiving and processing said AMR frames and said AMR control signal.

19. The network system according to claim 18, wherein said first and said second network element are separately arranged network elements.

20. The network system according to claim 18, wherein said third and said fourth network element are separately arranged network elements.

21. The network system according to claim 18, wherein said first and said second network element are arranged as a single network element.

22. The network system according to claim 18, wherein said third and said fourth network element are arranged as a single network element.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Gi
GGSN
Gn
SGSN
Iu-ps
RAN
BS
Uu
UE
CODEC
PORT# XX: RTP STREAM [AMR FRAMES]
PORT# YY: RTCP CONTROL [AMR CONTROL]
AMR FRAMES
AMR CONTROL

FIG. 1

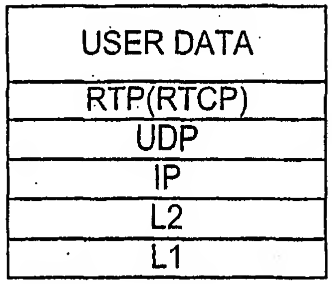
USER DATA
RTP(RTCP)
UDP
IP
L2
L1

FIG. 2

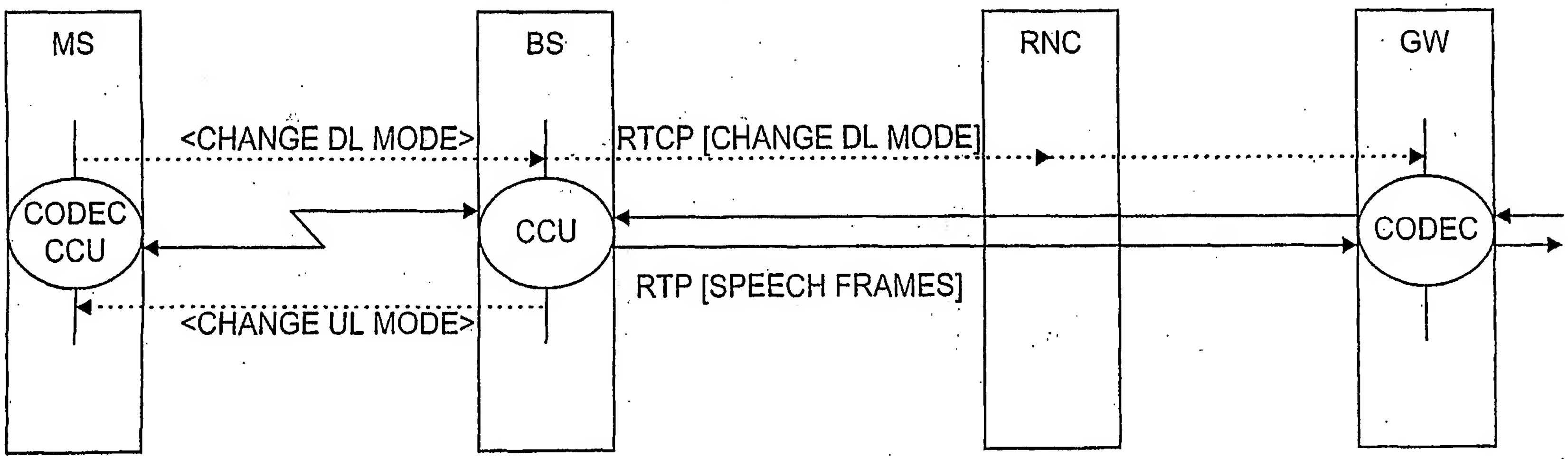
MS
BS
RNC
GW
CODEC
CCU
CCU
CODEC
<CHANGE DL MODE>
RTCP [CHANGE DL MODE]
<CHANGE UL MODE>
RTP [SPEECH FRAMES]

FIG. 3

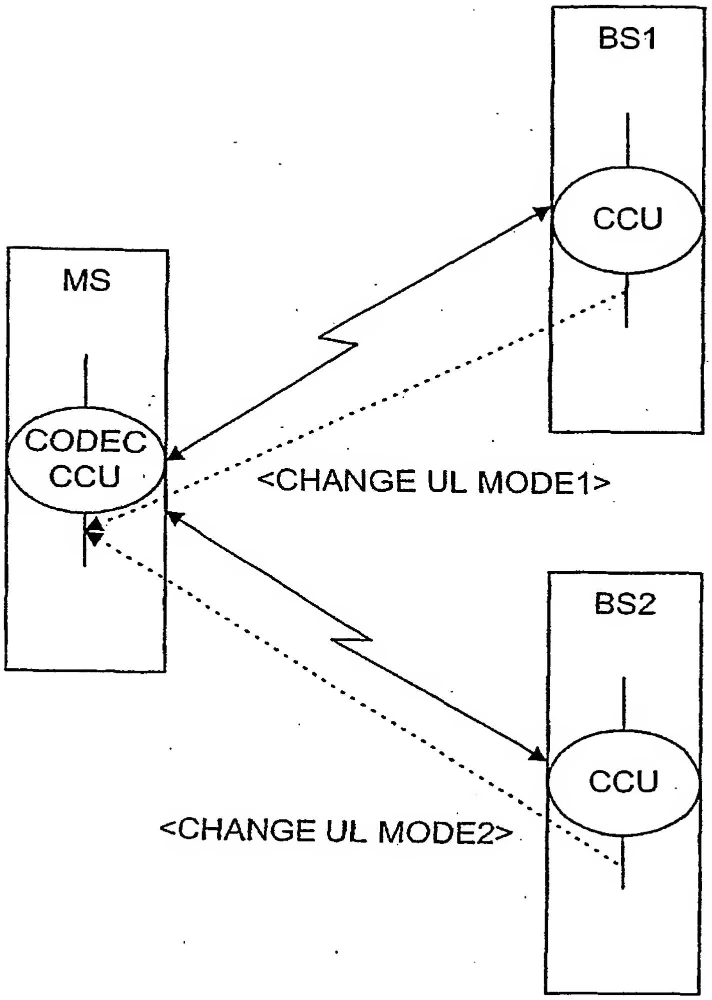

FIG. 4

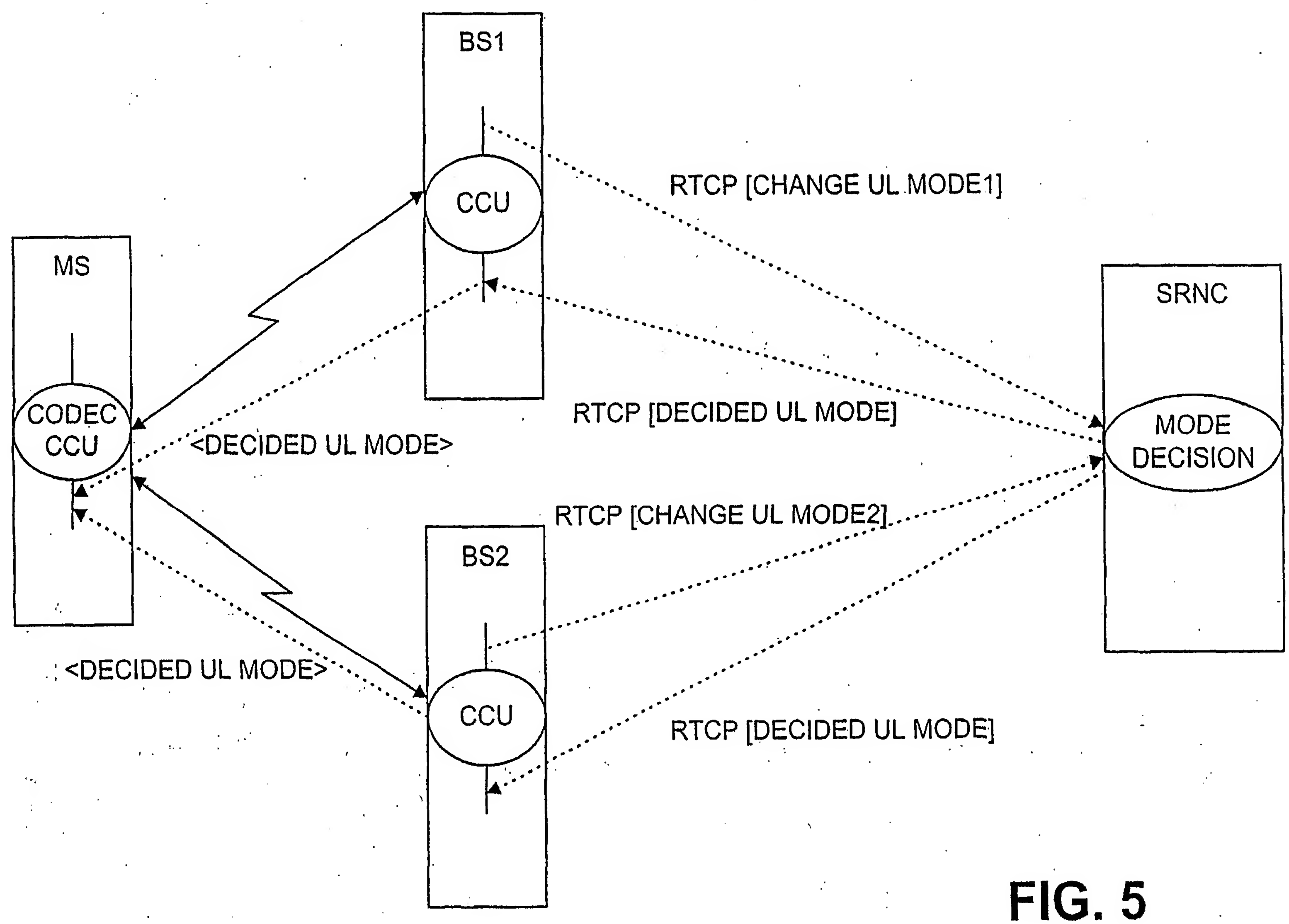
BS1
CCU
MS
CODEC
CCU
RTCP [CHANGE UL MODE1]
SRNC
MODE
DECISION
RTCP [DECIDED UL MODE]
<DECIDED UL MODE>
RTCP [CHANGE UL MODE2]
BS2
CCU
<DECIDED UL MODE>
RTCP [DECIDED UL MODE]

FIG. 5

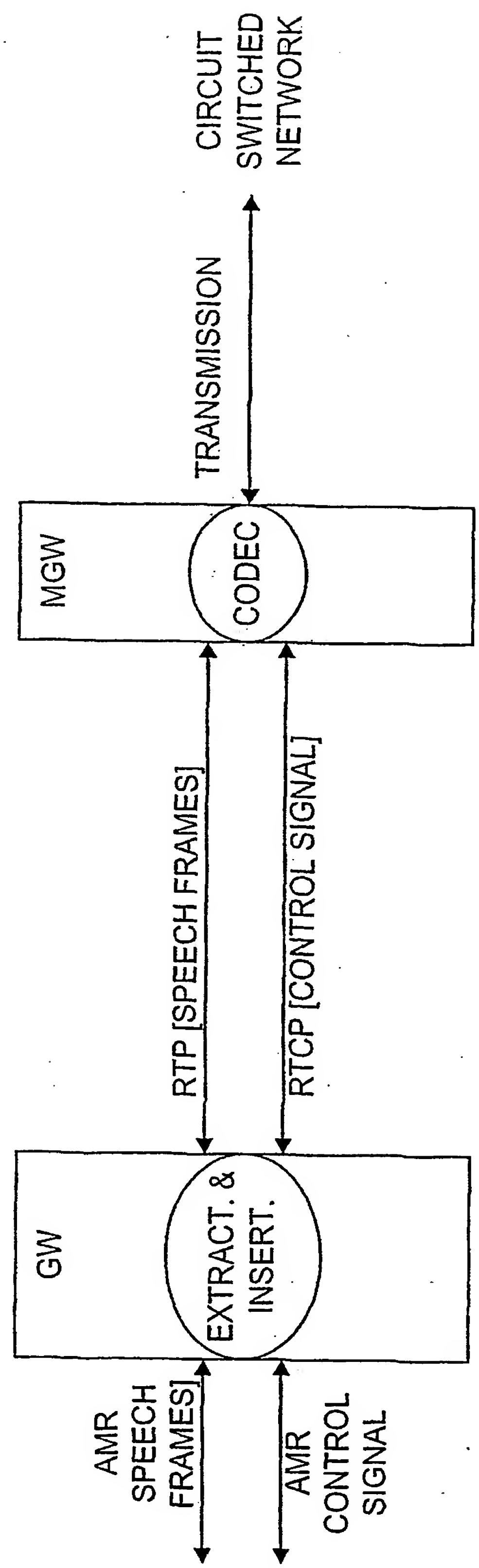
CIRCUIT
SWITCHED
NETWORK
TRANSMISSION
MGW
CODEC
RTP [SPEECH FRAMES]
RTCP [CONTROL SIGNAL]
GW
EXTRACT. &
INSERT.
AMR
SPEECH
FRAMES
AMR
CONTROL
SIGNAL

FIG. 6

INTERNATIONAL SEARCH REPORT

International Application No
PCT/EP 00/03233

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04Q7/30 H04Q7/38

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04Q H04M H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	BRUHN S ET AL: "CONCEPTS AND SOLUTIONS FOR LINK ADAPTATION AND INBAND SIGNALING FORTHE GSM AMR SPEECH CODING STANDARD" HOUSTON, TX, MAY 16 - 20, 1999,NEW YORK, NY: IEEE,US, vol. CONF. 49, 1999, pages 2451-2455, XP000922989 ISBN: 0-7803-5566-0	1,2,7-9, 11,18-22
A	abstract page 2451, left-hand column, line 15 -page 2452, left-hand column, line 29 page 2454, left-hand column, line 17 -right-hand column, line 21	3-6,10, 12-17
	-/--	

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
4 October 2000	07/11/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Rabe, M

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
Y	SISALEM D: "FAIRNESS OF ADAPTIVE MULTIMEDIA-APPLICATIONS" ATLANTA, GA, JUNE 7 - 11, 1998,NEW YORK, NY: IEEE,US, vol. CONF. 5, 7 June 1998 (1998-06-07), pages 891-895, XP000891000 ISBN: 0-7803-4789-7	1,2,7-9, 11,18-22
A	abstract page 891, right-hand column, line 19 - line 34	3-6,10, 12-17
A	GB 2 330 486 A (MOTOROLA LTD) 21 April 1999 (1999-04-21) the whole document	3-6, 13-15
A	US 5 768 308 A (RABIPOUR RAFI ET AL) 16 June 1998 (1998-06-16) abstract column 1, line 14 - line 25 column 1, line 53 -column 2, line 27 column 4, line 13 - line 62 figure 2	1-22
A	US 5 757 792 A (AOKI JINICHI) 26 May 1998 (1998-05-26) abstract column 1, line 48 - line 52 column 2, line 5 - line 35 figure 2	1,8,18
A	VILLETTE S ET AL: "SPLIT BAND LPC BASED ADAPTIVE MULTI-RATE GSM CANDIDATE" PHOENIX, AZ, MARCH 15 - 19, 1999,NEW YORK, NY: IEEE,US, 15 March 1999 (1999-03-15), pages 249-252, XP000900105 ISBN: 0-7803-5042-1 abstract page 249, left-hand column, line 16 - line 28 page 251, left-hand column, line 5 - line 29	1,8,18

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/EP 00/03233

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
GB 2330486	A	21-04-1999	NONE			
US 5768308	A	16-06-1998	CA	2207550	A	27-06-1996
			WO	9619907	A	27-06-1996
			CN	1170492	A	14-01-1998
			EP	0799554	A	08-10-1997
			JP	10500829	T	20-01-1998
US 5757792	A	26-05-1998	JP	2954000	B	27-09-1999
			JP	9065425	A	07-03-1997